IN THE CLAIMS

1. (Previously presented) A method for providing an audio stream in a voice over Internet Protocol (VoIP) environment, comprising:

determining a quality value for each of a plurality of audio streams communicated in a VoIP format;

selecting one of the audio streams based on the quality values for the audio streams; and

facilitating playing of the selected audio stream to a call on hold.

2. (Original) The method of Claim 1, wherein the audio stream comprises a music-on-hold channel from a music-on-hold server.

3. (Original) The method of Claim 1, wherein the quality value for an audio stream comprises at least one of packet jitter and packet loss for the audio stream.

4. (Original) The method of Claim 1, further comprising selecting the audio stream comprising a highest quality value.

5. (Original) The method of Claim 1, wherein the quality value for an audio stream comprises a current value for the audio stream determined based on real-time performance of the audio stream at a point at least proximate to a device playing the selected audio stream to the call on hold.

6. (Previously presented) The method of Claim 1, wherein the selected audio stream comprises a first audio stream, further comprising:

in response to at least degradation of the first audio stream below a threshold, selecting a second audio stream based on a then current quality value for each of the remaining audio streams; and

facilitating playing of the second audio stream.

7. (Original) The method of Claim 1, further comprising determining the quality value for each audio stream based on a sliding window of quality metrics for the audio stream.

8. (Previously presented) The method of Claim 6, wherein facilitating playing of the second audio stream comprises switching from the first audio stream to the second audio stream at an endpoint playing the first and second audio streams to the call on hold.

9. (Original) The method of Claim 1, further comprising presenting to users for selection only audio streams with a quality value above a threshold.

10. (Original) The method of Claim 1, wherein determining the quality value for each of the plurality of audio streams comprises:

receiving the plurality of audio streams; and

monitoring each of the quality streams based on at least one of packet jitter and packet loss of the audio stream.

11. (Previously presented) The method of Claim 1, wherein facilitating playing of the selected audio stream to a call on hold comprises communicating at least an identifier of the selected audio stream to an endpoint handling the call on hold.

12. (Original) The method of Claim 1, wherein determining the quality value for each of the plurality of audio streams comprises receiving the quality values for the audio streams from an upstream device in an Internet Protocol network.

13. (Original) The method of Claim 12, wherein the upstream device comprises an edge router of the Internet Protocol network.

14. (Previously presented) The method of Claim 1, further comprising:

selecting a locally stored audio file in response to at least the quality values for the audio streams being below a threshold value; and

facilitating playing of the stored audio file to a call on hold.

15. (Original) The method of Claim 1, further comprising receiving a list of audio streams, the plurality of audio streams including at least one of the audio streams identified by the list.

16. (Original) The method of Claim 15, wherein the list is generated by a call manager.

17. (Original) The method of Claim 1, further comprising generating a list of locally used audio streams, the plurality of audio streams including at least one of the locally used audio streams.

18. (Original) The method of Claim 1, further comprising:

identifying a poor quality audio stream based on the quality value for the audio stream; and

communicating an identifier of the poor quality stream to an upstream router for discard of the poor quality audio stream.

19. (Previously presented) A system for providing an audio stream in a voice over Internet Protocol (VoIP) environment, comprising:

means for determining a quality value for each of a plurality of audio streams communicated in a VoIP format;

means for selecting one of the audio streams based on the quality values for the audio streams; and

means for facilitating playing of the selected audio stream to a call on hold.

20. (Original) The system of Claim 19, wherein the audio stream comprises a music-on-hold channel from a music-on-hold server.

21. (Original) The system of Claim 19, wherein the quality value for an audio stream comprises at least one of packet jitter and packet loss for the audio stream.

22. (Original) The system of Claim 19, further comprising means for selecting the audio stream comprising a highest quality value.

23. (Original) The system of Claim 19, wherein the quality value for an audio stream comprises a current value for the audio stream determined based on real-time performance of the audio stream at a point at least proximate to a device playing the selected audio stream to the call on hold.

24. (Previously presented) The system of Claim 19, wherein the selected audio stream comprises a first audio stream, further comprising:

means for, in response to at least degradation of the first audio stream below a threshold, selecting a second audio stream based on a then current quality value for each of the remaining audio streams; and

means for facilitating playing of the second audio stream.

25. (Original) The system of Claim 19, further comprising means for determining the quality value for each audio stream based on a sliding window of quality metrics for the audio stream.

26. (Previously presented) The system of Claim 24, wherein facilitating playing of the second audio stream comprises means for switching from the first audio stream to the second audio stream at an endpoint playing the first and second audio streams to the call on hold.

27. (Original) The system of Claim 19, further comprising means for presenting to users for selection only audio streams with a quality value above a threshold.

28. (Original) The system of Claim 19, wherein the means for determining the quality value for each of the plurality of audio streams comprises:

means for receiving the plurality of audio streams; and

means for monitoring each of the quality streams based on at least one of packet jitter and packet loss of the audio stream.

29. (Previously presented) The system of Claim 19, wherein facilitating playing of the selected audio stream to a call on hold comprises communicating at least an identifier of the selected audio stream to an endpoint handling the call on hold.

30. (Original) The system of Claim 19, wherein the means for determining the quality value for each of the plurality of audio streams comprises means for receiving the quality values for the audio streams from an upstream device in an Internet Protocol network.

31. (Original) The system of Claim 30, wherein the upstream device comprises an edge router of the Internet Protocol network.

32. (Previously presented) The system of Claim 19, further comprising:

means for selecting a locally stored audio file in response to at least the quality values for the audio streams being below a threshold value; and

means for facilitating playing of the stored audio file to a call on hold.

33. (Original) The system of Claim 19, further comprising means for receiving a list of audio streams, the plurality of audio streams including at least one of the audio streams identified by the list.

34. (Original) The system of Claim 33, wherein the list is generated by a call manager.

35. (Original) The system of Claim 19, further comprising means for generating a list of locally used audio streams, the plurality of audio streams including at least one of the locally used audio streams.

36. (Original) The system of Claim 19, further comprising:

means for identifying a poor quality audio stream based on the quality value for the audio stream; and

means for communicating an identifier of the poor quality stream to an upstream router for discard of the poor quality audio stream.

37. (Previously presented) A system for providing an audio stream in a voice over Internet Protocol (VoIP) environment, the system comprising logic encoded in media and operable to:

determine a quality value for each of a plurality of audio streams communicated in a VoIP format;

select one of the audio streams based on the quality values for the audio streams; and

facilitate playing of the selected audio stream to a call on hold.

38. (Original) The system of Claim 37, wherein the audio stream comprises a music-on-hold channel from a music-on-hold server.

39. (Original) The system of Claim 37, wherein the quality value for an audio stream comprises at least one of packet jitter and packet loss for the audio stream.

40. (Original) The system of Claim 37, the logic further operable to select the audio stream comprising a highest quality value.

41. (Original) The system of Claim 37, wherein the quality value for an audio stream comprises a current value for the audio stream determined based on real-time performance of the audio stream at a point at least proximate to a device playing the selected audio steam to the call on hold.

42. (Previously presented) The system of Claim 37, wherein the selected audio stream comprises a first audio stream, the logic further operable to:

in response to at least degradation of the first audio stream below a threshold, select a second audio stream based on a then current quality value for each of the remaining audio streams; and

facilitate playing of the second audio stream.

43. (Original) The system of Claim 37, the logic operable to determine the quality value for each audio stream based on a sliding window of quality metrics for the audio stream.

44. (Previously presented) The system of Claim 42, wherein facilitating playing of the second audio stream comprises switching from the first audio stream to the second audio stream at an endpoint playing the first and second audio streams to the call on hold.

45. (Original) The system of Claim 37, the logic further operable to present to users for selection only audio streams with a quality value above a threshold.

46. (Original) The system of Claim 37, the logic operable to determine the quality value for each of the plurality of audio streams by:

receiving the plurality of audio streams; and

monitoring each of the quality streams based on at least one of packet jitter and packet loss of the audio stream.

47. (Previously presented) The system of Claim 37, wherein facilitating playing of the selected audio stream to a call on hold comprises communicating at least an identifier of the selected audio stream to an endpoint handling the call on hold.

48. (Currently amended) The system of Claim 37, ~~to~~ the logic operable to determine the quality value for each of the plurality of audio streams by receiving the quality values for the audio streams from an upstream device in an Internet Protocol network.

49. (Original) The system of Claim 48, wherein the upstream device comprises an edge router of the Internet Protocol network.

50. (Previously presented) The system of Claim 37, the logic further operable to:

select a locally stored audio file in response to at least the quality values for the audio streams being below a threshold value; and

facilitate playing of the stored audio file to a call on hold.

51. (Original) The system of Claim 37, the logic further operable to receive a list of audio streams, the plurality of audio streams including at least one of the audio streams identified by the list.

52. (Original) The system of Claim 51, wherein the list is generated by a call manager.

53. (Original) The system of Claim 37, the logic further operable to generate a list of locally used audio streams, the plurality of audio streams including at least one of the locally used audio streams.

54. (Original) The system of Claim 37, the logic further operable to:

identify a poor quality audio stream based on the quality value for the audio stream; and

communicate an identifier of the poor quality stream to an upstream router for discard of the poor quality audio stream.

55. (Original) A method for providing music-on-hold at an endpoint of an Internet Protocol network, comprising:

receiving a plurality of music-on-hold streams;

repetitively determining a real-time quality value for each of the audio streams based on at least one of packet jitter and packet loss for the music-on-hold stream;

in response to at least a call being placed on hold, selecting one of the music-on-hold streams as a high quality stream based on the real-time quality values for the music-on-hold streams; and

playing the high quality stream to the call on hold.

56. (Previously presented) The method of Claim 1, wherein facilitating playing of the selected audio stream to a call on hold comprises playing the selected audio stream at an endpoint.

57. (Previously presented) The system of Claim 19, wherein facilitating playing of the selected audio stream to a call on hold comprises playing the selected audio stream at an endpoint.

58. (Previously presented) The system of Claim 37, wherein facilitating playing of the selected audio stream to a call on hold comprises playing the selected audio stream at an endpoint.